

ELECTRONIC FUNDAMENTALS

SERVICE PRACTICES 28

PUBLIC ADDRESS SYSTEMS II

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Service Practices 28

INTRODUCTION

In this booklet, we take up use of commercial PA equipment. The electronic circuits for PA systems are explained in Service Practices 27.

A variety of commercial equipment is available. One kind differs from another in terms of the input devices that can be connected, the audio power that can be developed, and the speakers that can be connected. The kind of equipment you choose for a particular PA installation depends upon these factors. But before you choose PA equipment, you must know something about *acoustics*, the study of how sound behaves in the place where you intend to use the equipment.

This booklet covers acoustics, the use of typical commercial PA equipment, use of a tape recorder with PA equipment, and briefly, the service and maintenance of PA equipment.

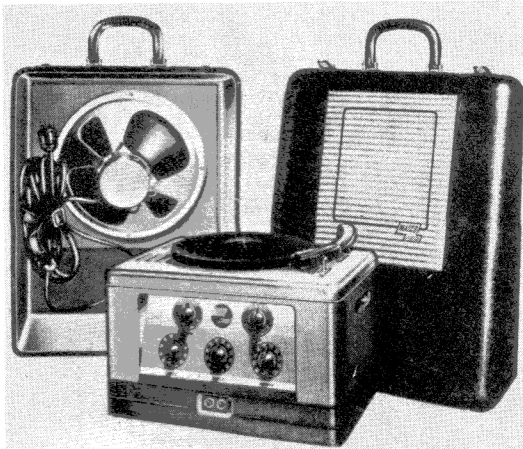
28-1. KINDS OF COMMERCIAL PA EQUIPMENT

Commercial PA equipment can be purchased in *package* units, including the amplifier, microphone, loudspeakers and, sometimes, a record player. Also, these components may be purchased separately. Low-power amplifiers are completely contained in a single cabinet and include a preamplifier-mixer as well as a power amplifier with both circuits mounted on the same chassis. High-power units are arranged in this way too. But sometimes high-power power amplifiers are made in separate units called *booster amplifiers* or *bridging amplifiers*. A very elaborate preamplifier-mixer

circuit — one with many mike and phono channels with separate controls for each — may be housed separately.

A typical portable package unit is shown in Fig. 28-1. The complete package is shown in (a). It consists of an amplifier with record player attached and two cone type loudspeakers. The two halves of the carrying case serve as baffles for the speakers. When the equipment is transported, the amplifier unit fits inside the carrying case. Figure 28-1b shows the package wrapped up and ready for transportation. The amplifier alone appears at c. It has input provision for two microphones in addition to the record player, which is permanently connected to the amplifier input. The gain of each mike channel and the phono channel can be separately controlled. In addition, the amplifier has separate bass and treble tone controls, which affect all three input channels at once. That is, they control the tone of the combined audio power output. The maximum power output of this amplifier is 17 watts.

A similar amplifier made by another manufacturer is shown in Fig. 28-2. This one has a maximum power output of 30 watts. Like the 17-watt amplifier, it has provision for two microphone inputs and one phonograph input. This time, the phono is not built into the amplifier. The three input channels can be controlled separately. The amplifier has a bass control and a treble tone control. Both of the amplifiers shown in Fig. 28-1 and 28-2 have illuminated control panels — a feature of many modern PA amplifiers. These two amplifiers are typical of low-to-medium power self-contained units in which the preamplifier and the power amplifier are housed together.



(a)



(b)



(c)

Fig. 28-1



Fig. 28-2

A PA system centering around one of these amplifiers can be set up as shown in diagram form in Fig. 28-3. The amplifier puts out 30 watts and drives two speakers, each drawing 15 watts.

Suppose we want to add two more speakers to the arrangement shown in the figure. Each of the new speakers draws 15 watts. Adding the two speakers, which together draw 30 watts, to the original two speakers, which together draw 30 watts, makes the total power load 60 watts. Our amplifier is capable of delivering only 30 watts. Therefore, we need an additional amplifier.

We can connect the second amplifier, which is identical to the first, as shown in Fig. 28-4. We run a lead from the output speaker line of amplifier No. 1 to the phono input terminal of amplifier No. 2. Since the phono input terminal is a high-impedance input, it draws almost no power from amplifier No. 1. However, the output signal voltage from amplifier No. 1 is applied to the phono input of amplifier No. 2. As you have learned, voltage alone, and not power, is what is needed to drive an amplifier.

The two additional speakers are connected to the output of amplifier No. 2 as shown in

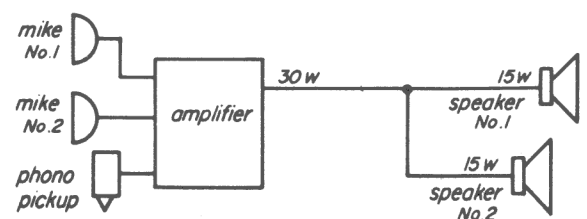


Fig. 28-3

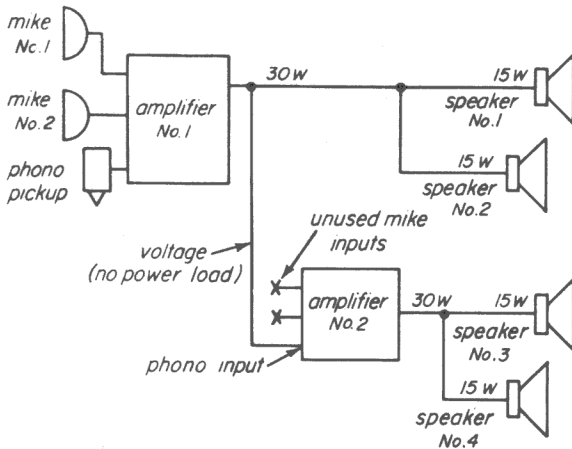


Fig. 28-4

Fig. 28-4. By means of the arrangement shown, all sound signals that enter amplifier No. 1 are reproduced in all four speakers.

In the arrangement in Fig. 28-4, we want to have all sounds that enter the system come out of all four speakers. But if we connect microphones to the mike inputs of amplifier No. 2, the sound signals from the mikes, so connected, would be heard only from speakers No. 3 and No. 4. Therefore, the two microphone input channels of amplifier No. 2 are wasted. However, in actual practice, the amplifier used in the same position as amplifier No. 2 in Fig. 28-4 would not have wasted input terminals. Instead, a special commercial amplifier is used. Such commercial amplifiers called *booster* or *bridging* amplifiers have no mike channels, only a high-impedance/high-voltage level input terminal similar to the phono-input terminals of self-contained amplifiers. (Booster amplifiers with 500-ohm input terminals are also available. These too have only one input, and are intended for connection to low-voltage level lines, as we shall see a little later.)

A 125-watt commercial booster amplifier is shown in Fig. 28-5. In addition to not requiring mike inputs, it does not need a whole set of controls. It has only one input, so it needs only one volume control. This control adjusts its power output.

Figure 28-6 is a diagram showing how a booster is used with an amplifier like the 30-watt model shown in Fig. 28-2.

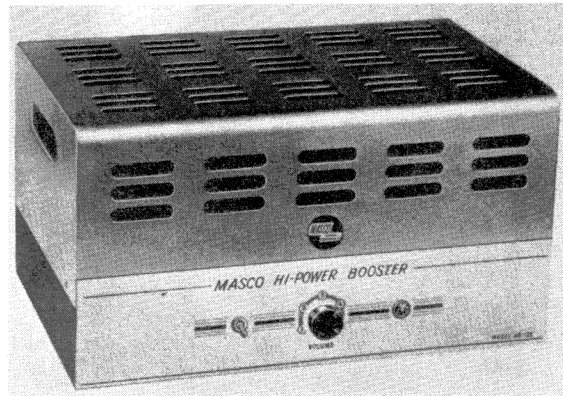


Fig. 28-5

The 30-watt amplifier has a separate volume control for each mike and phono channel as well as a tone control for the combined signal. The 125-watt booster amplifier simply boosts the power output of the combined signal. Therefore, the only control it has is its single volume control. The amplifier is called a *booster amplifier* because of this power boosting function. It can also be called a *bridging amplifier* because amplifiers like the 125-watt model are bridged across the speaker line of amplifiers like the 30-watt model without consuming any power from the speaker line.

Any number of bridging amplifiers may be connected to a speaker line. Because the input impedance of bridging amplifiers is high, they do not in themselves, impose a load upon the power amplifier driving the speaker line. Since some power amplifiers *must* be loaded, whenever you connect power amplifiers, including bridging amplifiers to the output terminals of a power amplifier, connect a speaker load or equivalent dummy-load resistor. In certain cases, the power amplifier output tubes can be damaged if no load is in place. In other cases, extreme distortion of the output voltage signal results. Yet, there are power amplifiers that need not be loaded. In general, the more negative feedback an amplifier has, the less is the need for loading.

Before we end our discussion of Fig. 28-6, there is one last comment to be made. The 125 watts output of the booster is distributed among six speakers. Three of these

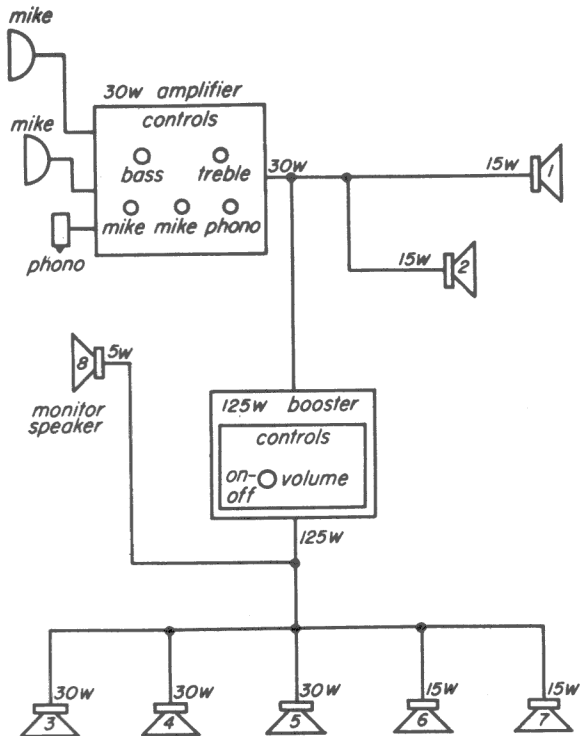


Fig. 28-6

draw 30 watts each and two draw 15 watts each. This makes a total of 120 watts. The remaining 5 watts is applied to a small speaker located near the 30 watt amplifier control board. The small speaker serves to monitor (check up on) the system. If either of the amplifiers becomes defective, the defect can be heard by someone near the control board. It is better to have the monitor speaker work off the booster output than to have it work off the output of the 30-watt amplifier because, working off the booster, the monitor will show a failure to the system operator whether the failure occurs in the booster or the 30-watt amplifier. If the monitor worked off the 30-watt amplifier, it would not show up a failure occurring in the booster.

Sometimes, a PA system is made up of a preamplifier-mixer unit driving one or more boosters. In such a system, the boosters supply all speaker power. The preamplifier is a voltage amplifier, the power output of which is not sufficient to operate even one small speaker. Its function is to provide audio voltage output to drive the boosters.

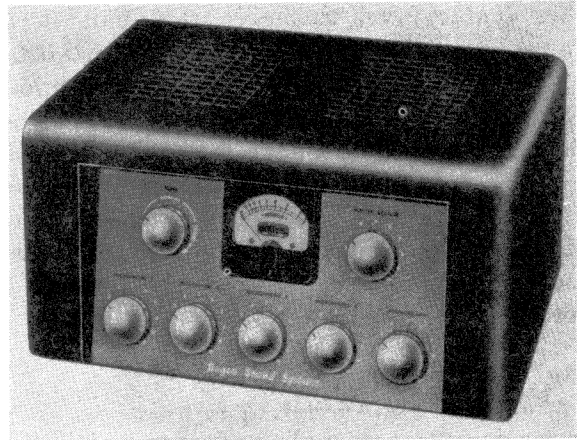


Fig. 28-7

A commercial preamplifier unit is shown in Fig. 28-7. The manufacturer gives the specifications for this model listed in Table A.

TABLE A – BOGEN PREAMPLIFIER SPECIFICATIONS

Output:	8 volts at less than 1% distortion into 10,000 ohms. 1 mw at less than 1% distortion into 500 ohms. 20 mw at less than 3½% distortion into 500 ohms.
Frequency Response:	±2 db from 20 to 15,000 cycles
Gain:	Microphone channels (4) high-impedance inputs: 78 db; phono (1) high-impedance: 36 db.
Hum:	–65 dbm
Input Impedance:	All high-impedance inputs: ½ megohm. Plug-in transformers for low-impedance of 50 or 200 or 500 ohms.
Power Consumption:	27 watts, 117 volts, 50–60 cycles a.c.
Tubes:	
Total 7:	5 – 6SF5, 1 – 6SN7, 1 – 6X5GT
Dimensions:	15¼" wide, 8" high, 11" deep.
Shipping Weight:	19 lbs.

The specifications tell you that the output line (cable) that connects this unit to a power amplifier can either be at a 10,000-ohm impedance level or at a 500-ohm impedance level. The output signal voltage for the 10,000-ohm impedance level is specified as 8 volts. It is conventional to give the signal level at low impedance (500 ohms) in milliwatts rather than volts, so the manufacturer gives the output into 500 ohms as 1 mw. From the formula

$$P = \frac{E^2}{R}$$

the signal voltage level at 500 ohms can be computed as follows:

$$P = \frac{E^2}{R}$$

$$0.001 = \frac{E^2}{500}$$

$$E^2 = 500 \times 0.001$$

$$= 0.5$$

$$E = \sqrt{0.5}$$

$$= 0.7 \text{ volt (approx)}$$

The voltage output at 500 ohms is less than 10 percent of the voltage output at 10,000 ohms. Since more signal voltage is available at the 10,000-ohm level, a power amplifier with lower sensitivity can be driven at the 10,000-ohm level. For this reason, use of the 10,000-ohm level is preferred except where a long connecting cable is used between the preamplifier and the power amplifier. Then the lower line impedance (500 ohms) results in less hum and noise pickup on the line. For instance, radio broadcast stations sometimes use a preamplifier like the one in Fig. 28-7 for remote pickup of a radio program. They connect the output of the unit to a rented 500-ohm telephone line that may be several miles long. At the other end of the line, in the broadcast

studio, they connect a booster amplifier. The booster they connect to the line must have a 500-ohm low impedance input. Of course, a booster to be connected to the 500-ohm output of a preamplifier must always have a 500-ohm rated input impedance whether it is used in a radio broadcast installation or a PA installation. Boosters with low-impedance inputs (500 ohms or other values) are the same in all respects as the boosters with high-impedance input that we discussed previously except that there is a step-up transformer at the low impedance booster input terminal. The transformer is needed to step up the signal voltage present on the low-impedance line. As we pointed out, the signal voltage delivered to a low-impedance line by a preamplifier is low. To sum up, we can say that low-impedance preamplifier output and booster input is to be used when a long cable connects them and that the signal voltage level will then be a low value. High-impedance preamplifier output and booster input is to be used when a relatively short cable connects them and then the signal voltage level will be a high value.

The high-impedance output of the preamplifier shown in Fig. 28-7, is 10,000 ohms. The high-impedance input of booster amplifiers is approximately equivalent to the phono input impedance of PA amplifier units, or about $\frac{1}{2}$ megohm. Nevertheless, the 10,000-ohm output of a preamplifier can be directly connected to the $\frac{1}{2}$ -megohm input of a booster. It is not necessary to match the output impedance of a preamplifier unit. In general, it is not necessary to match the output impedance of any voltage amplifier when connecting another piece of equipment to its output terminals. However, no device with an input impedance lower than the output impedance of the preamplifier should be connected because doing so loads the voltage amplifier tube excessively, reduces the output signal value, and reduces the maximum possible undistorted output voltage.

Even when a booster with a 500-ohm rated input impedance is connected to the 500 ohm output of a preamplifier, the im-

impedance is not actually being matched. This is because the actual impedance across the terminals of a booster (or any power amplifier) that is rated at 500 ohms is actually much higher than 500 ohms. It might be ten times as high as 500 ohms or 5,000 ohms. (This was explained in Service Practices 27 in connection with low-impedance microphone input to power amplifiers.) The 500-ohm input rating is just a conventional way the manufacturer has of saying that *you can connect a 500-ohm low signal voltage line to these terminals and be sure that there is a transformer here that will step up the signal voltage sufficiently to drive this amplifier.*

You may wonder why the manufacturer does not make the 500-ohm rated input actually equal 500 ohms and be done with all this uncertainty. It is better to have the 500-ohm booster input higher than 500 ohms. If it is ten times 500 ohms, up to ten boosters can be connected in parallel across a 500-ohm preamplifier output before the actual load on the preamplifier totals 500 ohms. If the eleventh booster is connected in this case, the actual load will be lower than 500 ohms and, as we pointed out earlier, it is bad to terminate a preamplifier in an impedance lower than its rated output impedance. In general, the higher the actual impedance of a 500-ohm booster input, the more of these boosters can be shunted across a 500-ohm line.

The output of the preamplifier of Fig. 28-7 is 1 milliwatt at 500 ohms. The preamplifier should be connected to a power amplifier or booster capable of full power output with an input of 1 milliwatt at 500 ohms. One milliwatt is zero dbm, so a one milliwatt level is sometimes called the zero level. The preamplifier is said to have a zero level output and the booster to be used with it is said to have a zero level input.

Let's compare the factors that have to do with using a booster connected to the output of another power amplifier with factors that apply when a preamplifier is used. Both power amplifiers of the kind found in self-contained PA units and booster amplifiers

have low-impedance output terminals. The output from these terminals is the same as that from the low-impedance terminals of preamplifier units as far as impedance is concerned, i.e., both may have a 500-ohm output. But the level of signal voltage at the 500-ohm output terminals of a power amplifier is much higher than the level of signal voltage at the 500-ohm output terminals of a preamplifier. We have seen that the signal voltage level at the 500-ohm terminals of the preamplifier of Fig. 28-7 is 0.7 volts when the preamplifier is delivering 1 mw. Compare this 0.7-volt value to the 124-volt signal voltage level when 30 watts are being delivered at the 500-ohm output terminals of a 30 watt power amplifier.

You have learned that the 500-ohm input rating of an amplifier tells you that *you can connect a 500-ohm low signal voltage line to these terminals and be sure that there is a transformer here that will step up the signal voltage sufficiently to drive this amplifier.* Obviously, the 124-volts delivered to 500 ohms by a 30 watt power amplifier is not a low signal voltage and does not require the services of a transformer to step up the signal voltage, so there is no need to bridge the speaker line with a 500-ohm booster. Instead, a high-impedance booster can be used in the manner described in connection with Fig. 28-6. However, a 500-ohm input is often designed to be balanced to ground while high-impedance inputs are usually single ended. Sometimes, a 500-ohm input device is used where the voltage step-up it provides is not wanted in order to take advantage of the balance feature.

You should avoid an error that has trapped many servicemen. Do not interpret the 500-ohm input-impedance rating of an amplifier to be the same as the 500-ohm tap on a speaker-to-line matching transformer. You will be making a mistake if you interpret these two ratings to mean the same thing. When a speaker is matched to a 500-ohm line by connecting the line to the 500-ohm tap on a speaker-to-line transformer with the speaker connected to the other side of the transformer, a 500-ohm load is imposed upon

the output of the amplifier feeding the 500-ohm line. But when the 500-ohm line is connected to the 500-ohm input terminals of a booster amplifier, no such load is imposed upon the amplifier feeding the line, because the actual input impedance of the booster is not 500 ohms at all; it is at least 5,000 ohms. At any rate, a 500-ohm booster need not be used to bridge a 500-ohm speaker line.

Now you probably realize that if the 500-ohm rated input impedance of an amplifier is not actually 500 ohms, it cannot give a 500-ohm impedance to a cable. That a cable has no impedance of its own and must be given an impedance by some device connected to it was brought out in Service Practices 26. The cable is given its 500-ohm impedance by the device connected at the beginning of the cable and not the amplifier connected at the end of the cable. If the beginning of a cable is connected to a microphone, the microphone gives the cable its impedance. If the beginning of the cable is connected to the output of a preamplifier, the preamplifier gives the cable its impedance. This is true whether the impedance is high or low. A cable that begins at the 10,000-ohm output of the preamplifier shown in Fig. 28-7 has an impedance of 10,000 ohms.

Now to get back to our PA system made up of a preamplifier driving a booster. The diagram of such a system is shown in Fig. 28-8. We are assuming that the system is installed in an outdoor theatre. The equipment used is as follows:

Five 100-watt speakers with 70-volt constant voltage inputs.

One 15-watt monitor speaker.

Five 125-watt boosters with zero level input and 70-volt constant-voltage output.

Four low-impedance microphones with 200-ohms output.

Four 200 ohm-to-high-impedance transformers for matching the mikes to the amplifier.

One preamplifier with zero level output: high-impedance mike inputs and one high-impedance phono input.

One phonograph.

The preamplifier is like the one shown in Fig. 28-7. The boosters are like the one shown in Fig. 28-5. Low impedance mikes are used because of the great length of cable between mike and preamplifier input. Because the preamplifier microphone inputs are normally high, plug-in transformers must be used with the preamplifier to match the 200-ohm impedance of the mike lines—that is, to step up the low signal voltage on these lines so that it will equal the voltage normally delivered by a high-impedance mike. Therefore, the voltage will be suitable for application to the high-impedance mike input terminals of the preamplifier.

The proportion of sound from each mike that is reproduced in the speakers can be controlled in the control booth by means of the mike gain controls on the preamplifier. If it is necessary to shut off any or all of the mikes, this can be done by setting the gain control at minimum. Sound from the phonograph can be fed into the system at any time and can be controlled at the preamplifier. Whoever operates the controls on the preamplifier knows exactly what he is doing because he can hear in the monitor speaker the result of twisting any of the knobs on the preamplifier. The power delivered to each individual speaker can be set at 100 watts or any lower value by means of the volume control on the booster driving each speaker. Since 70-volt constant voltage lines are used between booster and speaker, there is no impedance-matching problem involved in connecting the monitor to the output of the last booster. All that need be considered here is power. The booster can deliver 125 watts and so it is all right to connect a 15-watt speaker in addition to the 100-watt speakers to draw a total of 115 watts.

Amplifiers, preamplifiers, boosters, radio tuning units, record players, and etc. are

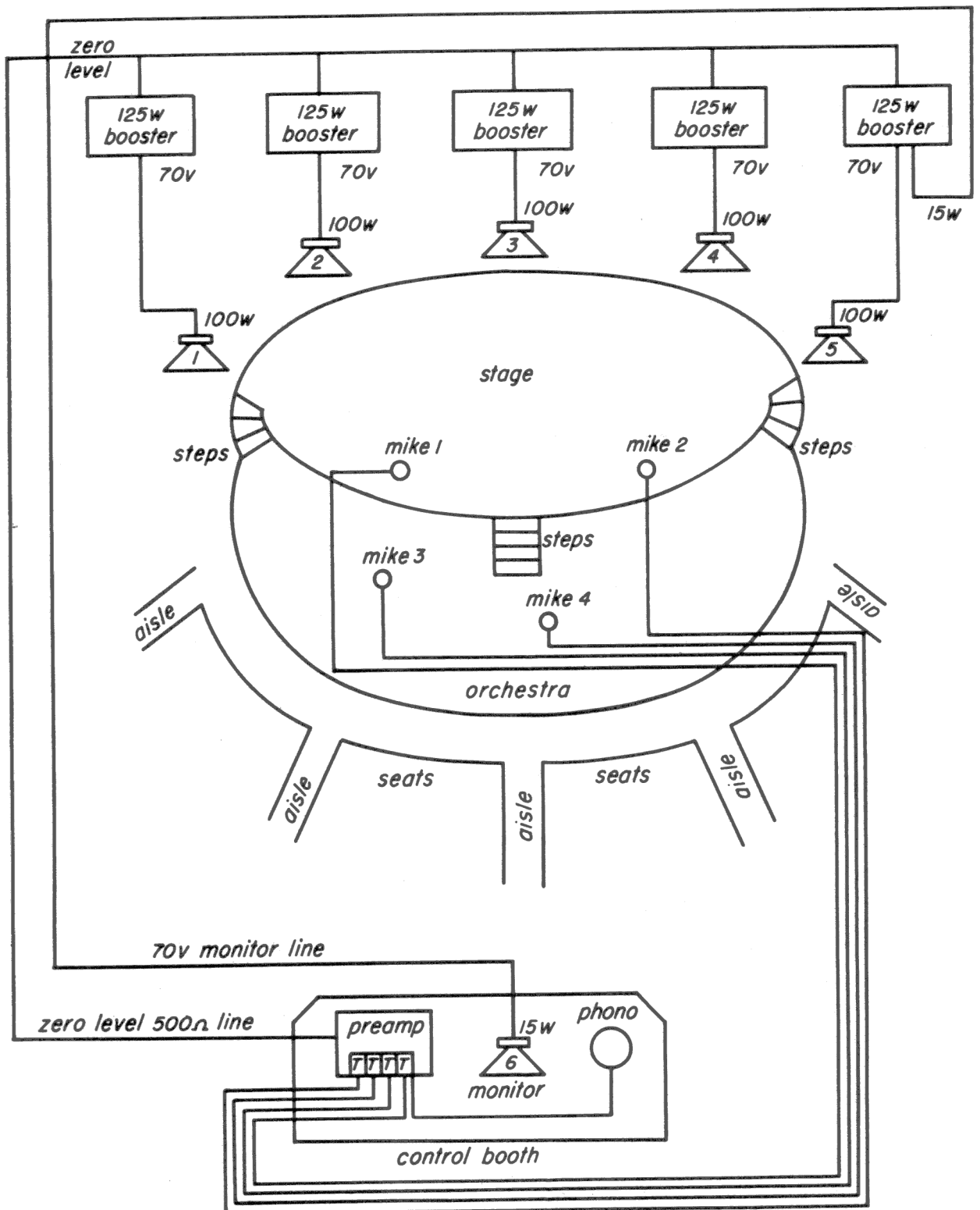


Fig. 28-8

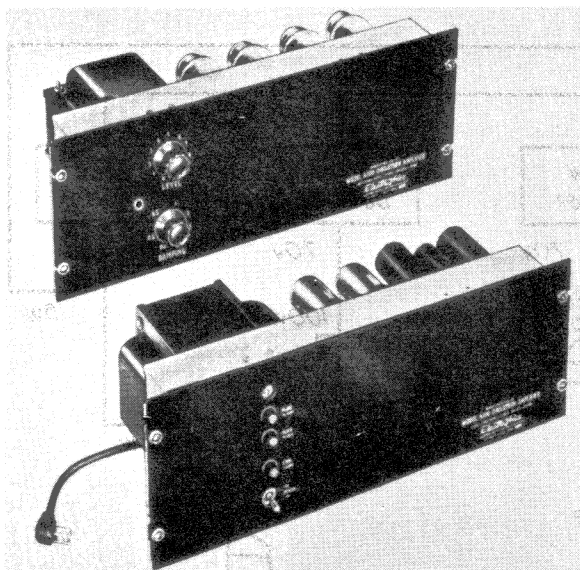


Fig. 28-9

available in a form suitable for rack mounting. Rack-mounted units are not housed in a cabinet designed to rest on a table or other flat surface. Instead, the chassis is attached to a flat metal plate. Figure 28-9 shows the amplifier and power supply chassis of the Electro-Voice Circlotron® circuit. The chassis are attached to flat plates that can be screwed to the flange of tall metal cabinets designed for housing rack-mounted equipment. Examples of such cabinets may be seen in Fig. 28-10 and 28-11. Figure 28-10 shows a set of rack-mounted PA units made by the David Bogen Co. Inc., of New York City and Fig. 28-11 shows a set of Masco® rack-mounted PA equipment made by the Mark Simpson Manufacturing Co. Inc., of Long Island City, N.Y. Where a large number of units are to be housed together, the rack mounting arrangement is neat and convenient. Blank plates are installed in the spaces in the cabinet where no chassis is mounted. Whenever a new chassis has to be added, it replaces the blank plate. This way of mounting PA equipment is used extensively in PA installations used in factories for paging and announcing purposes and for piping music to the work areas.

28-2. SPEAKERS

Types. Various kinds of loudspeakers are used in PA installations. The cone type speaker (Fig. 28-12a) is used in indoor in-

stallations. Audio power up to 50 watts can be applied to a cone type speaker. Cone speakers are usually mounted in baffles like the one pictured in Fig. 28-12b.

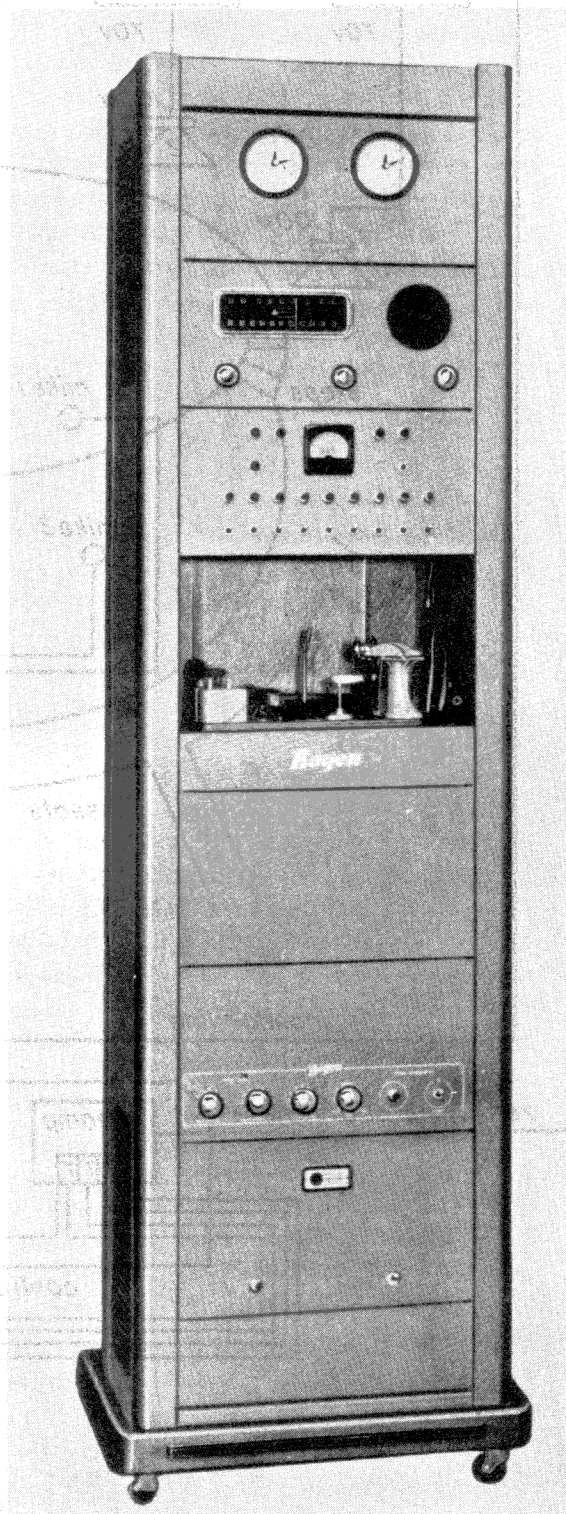
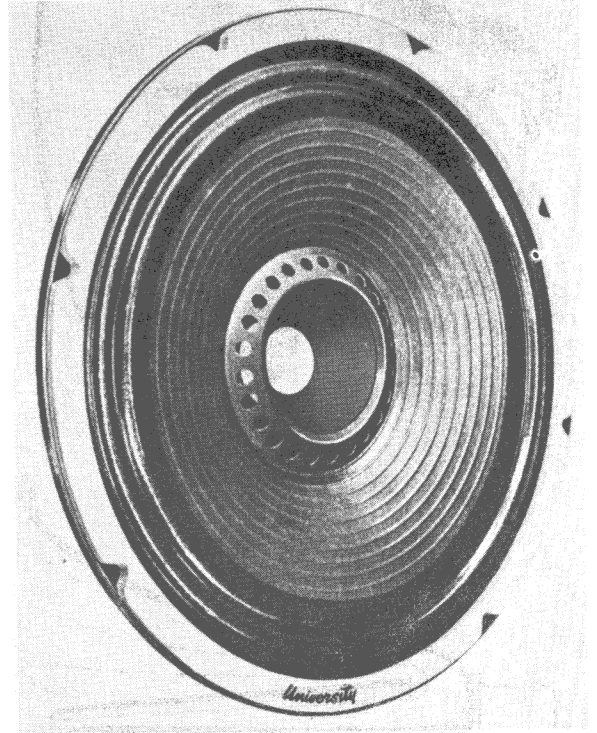


Fig. 28-10

Cone speakers are not used out-of-doors because the fibre cone cannot withstand exposure to the weather. For outdoor PA installations, horn-type loudspeakers are used. Figure 28-13a shows a folded horn made by the Jensen Radio Manufacturing Co. of Chicago, Ill. Figure 28-13b shows how sound waves travel through a folded horn. The sound originates in a detachable *driver unit* or *speaker motor* like that shown in Fig. 28-14. Figure 28-15 shows a folded horn with the driver unit clearly visible in the rear. This assembly is made by University Loudspeakers, Inc. of White Plains, N.Y. Figure 28-16 shows two horns driven by a single driver unit.

Since horn-type speakers are made of metal and have no large exposed fibre cone, they can be used outdoors. However, they have other advantages as well, and they are often used in indoor PA systems. Horn-type

speakers are much more efficient than cone type speakers. Much less electric power is required to get a given loudness of sound from a horn than is required to get the same volume from a cone type speaker. Also, horns radiate sound in a beam that can be aimed at an area where it is needed most.



(a)

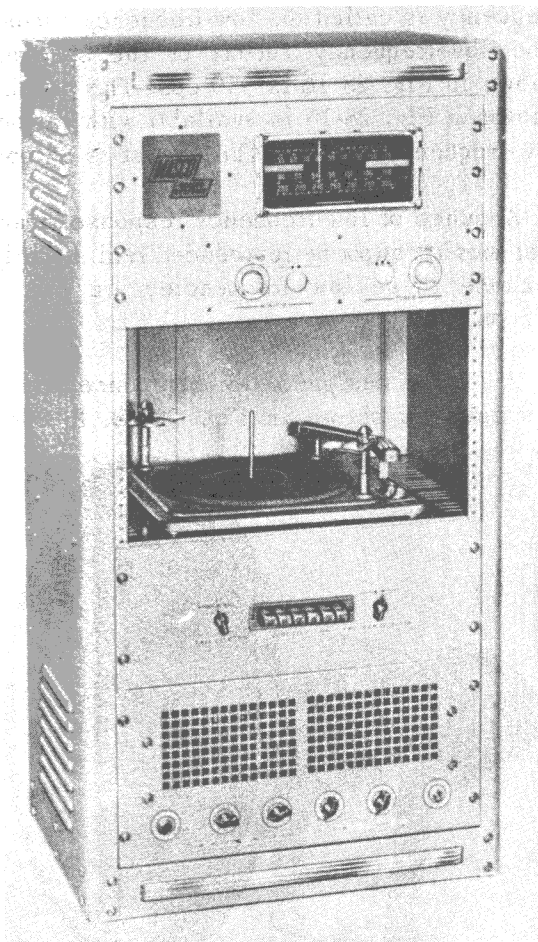
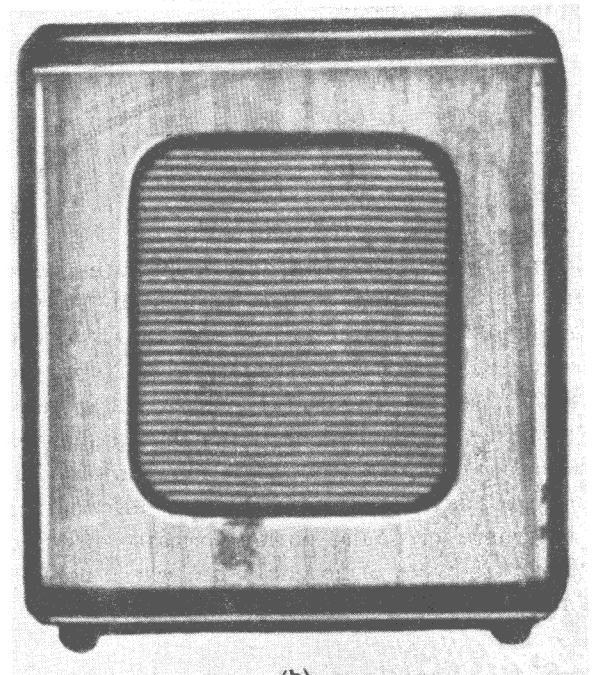


Fig. 28-11

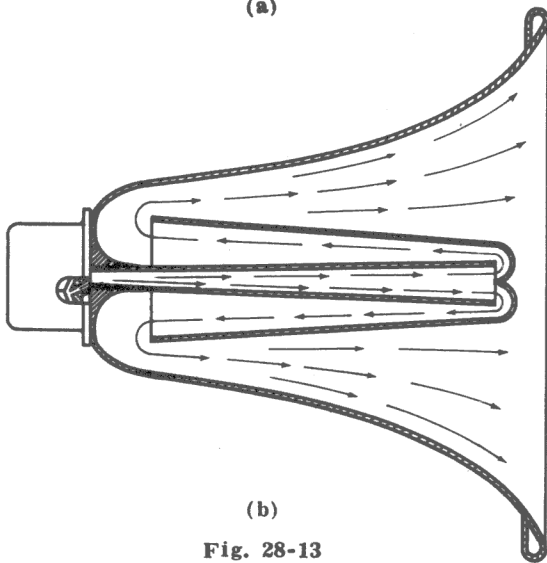


(b)

Fig. 28-12



(a)



(b)

Fig. 28-13

Horn speakers are available to radiate sound in various beam widths measured by the angle of dispersion in degrees. Each of the speakers in Fig. 28-16 radiates a 120 degree beam of sound. Figure 28-17 shows a speaker (made by the University Co.) that disperses sound at a 120-degree angle horizontally but only 60 degrees vertically, thus eliminating the waste of sound power normally directed toward the roof of an enclosure where nobody is in position to hear it.

The principle disadvantage of horns is that a very large one is required to reproduce very low audio frequencies. As a result, the

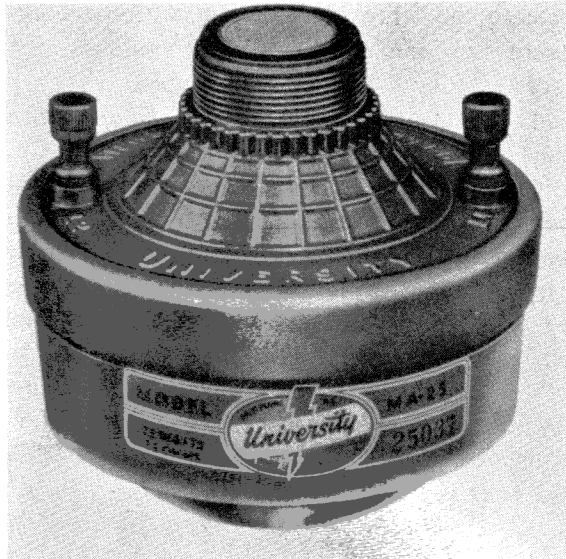


Fig. 28-14

run-of-the-mill horn does not reproduce frequencies as low as those reproduced by cone-type speakers. Horns are rated in terms of the lowest frequency they reproduce. This frequency is called the low-frequency cut-off. The low-frequency cut-off of the speakers shown in Fig. 28-16 is 350 cps. The speaker shown in Fig. 28-15 is available with various low-frequency cutoffs. The lowest is 85 cps.

Absence of low-frequency response means that music cannot be reproduced realistically, because these low frequencies are part of the music.

A horn designed to reproduce music faithfully is shown in Fig. 28-18. It has a



Fig. 28-15

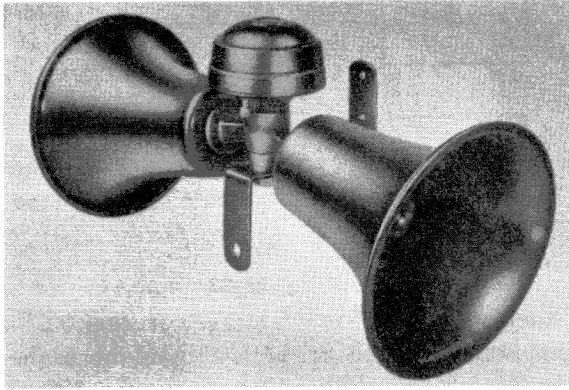


Fig. 28-16

frequency response of 50 to 15,000 cps which is very good. Because of its excellent low-frequency response, it must be large. It has a diameter of $33\frac{1}{2}$ inches and weighs 80 pounds. Absence of low-frequency response is an advantage, rather than a disadvantage in the reproduction of speech and even of music in noisy locations. This point will be covered more fully when we take up acoustics.

Two interesting special-purpose horns are shown in Fig. 28-19 and Fig. 28-20. The speaker in Fig. 28-19 is waterproof. It will operate under water. It is used on ships and in moist locations like a boiler room. The speaker shown in Fig. 28-20 is explosion-proof. It is used in locations where gas, dust, or vapor are present. An electric spark from an ordinary speaker might cause an explosion under these conditions. In the explosion-proof speaker, the electrical mechanism is completely sealed off and protected.

Probably, the principle advantages of horns are their efficiency and ability to handle high power. Some horns can handle

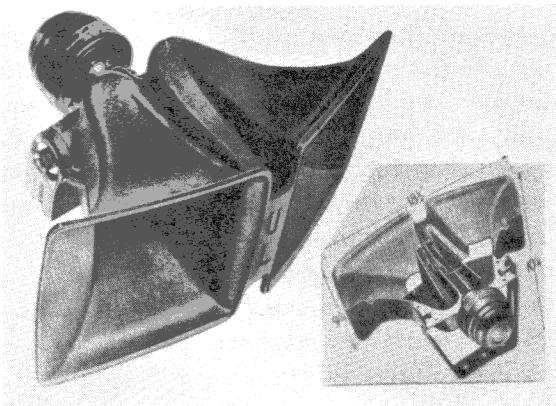


Fig. 28-17

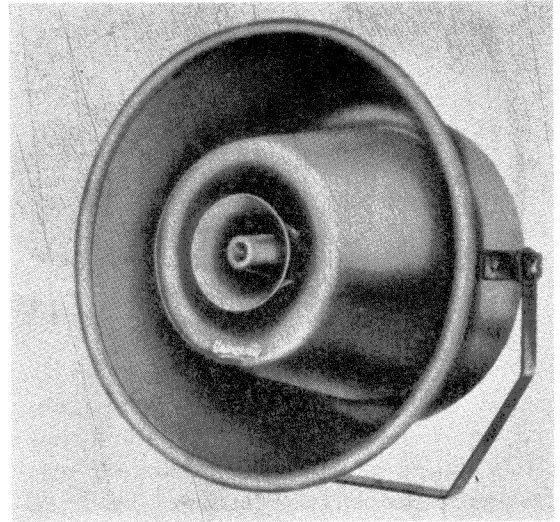


Fig. 28-18

600 watts and more. These horns are made up of several driver units driving a single horn. You may have heard church chimes amplified and sent through such a system.

Speaker Overload. A speaker is overloaded when it is caused to consume more electric power than it is rated to handle. In addition, horn-type speakers may be overloaded if they are caused to consume smaller amounts of electric power at a frequency below the low-frequency cutoff of the horn. The design of horns is such that the air offers no opposition to the motion of the diaphragm in the driver unit at frequencies below cutoff. The low-

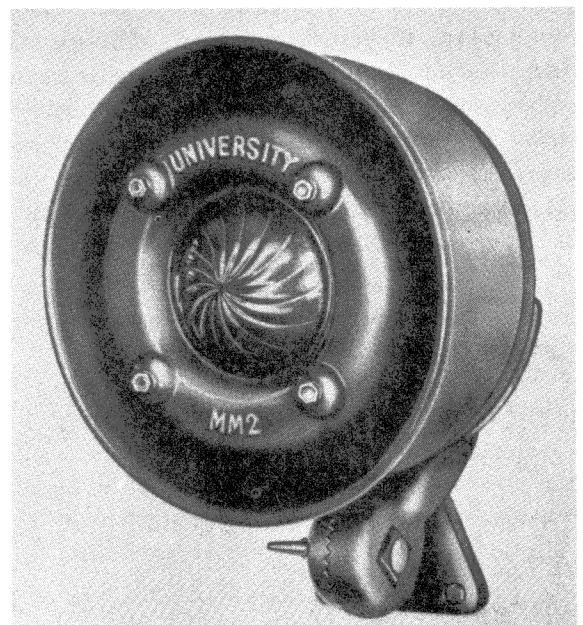


Fig. 28-19



Fig. 28-20

frequency cutoff is determined by the shape and size of the horn and has nothing to do with the driver unit. For this reason, electric filters are sometimes used in PA installations with horn speakers to prevent the low audio frequencies from reaching the horn driver unit. The filter may be nothing more than a capacitor of proper value in series with the speaker load (Fig. 28-21). Capacitors have a higher impedance (reactance) at low frequencies than at high frequencies. The capacitor in Fig. 28-21 is in a position to block out low frequencies. The value is chosen so that the low frequencies will be attenuated sufficiently while the highs are not attenuated too much. The appropriate value depends upon the impedance of the load connected to the end of the line, which in the case of speaker loads is equal to the line impedance. (Where the line load is an amplifier, the actual impedance of the amplifier as a load does not equal the line impedance, even though the amplifier rated impedance does equal the line impedance.) A value of 0.05 microfarads is appropriate for use with 500-ohm speaker

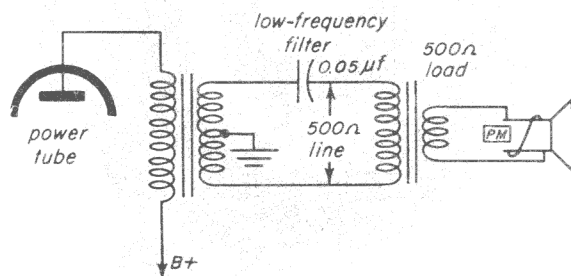


Fig. 28-21

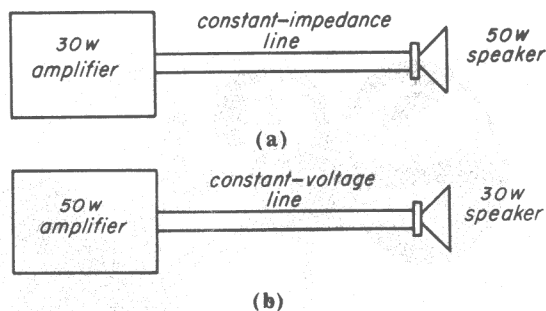


Fig. 28-22

lines. The working voltage of the 0.05- μ f capacitor should be higher than the highest audio-voltage peak to occur on the 500-ohm line. This, in turn, depends upon how much audio power is carried on the line. A value of 600 wv is usually safe.

Speaker lines at voice-coil impedance levels (8 to 16 ohms) require a series capacitor of about 30 μ f. The capacitor cannot be a conventional electrolytic filter capacitor. Such capacitors, being polarized, are useful only in d-c circuits. The audio signal on a speaker line is a.c. Special a-c electrolytic capacitors may be used. These are available commercially. They are used for many purposes in addition to the one we are describing. For instance, they are used as *starting capacitor* in connection with a-c electric motors and as *power-factor correctors* in modern window unit air conditioners.

All speakers in PA systems should be protected against overload. The most positive way of doing this is to make sure that too much power can never be delivered to a speaker. When a constant-impedance speaker-power distribution system is in use, this means that the speaker load must be rated to handle more power than the amplifier is capable of delivering to it, as illustrated in Fig. 28-22a. This is so because, in the constant-impedance system, the full power of the amplifier can always be delivered to the speaker load by advancing the volume control. However, in the constant-voltage system, the speaker can never draw more than its rated power, provided the system was properly set up initially with the amplifier set to deliver maximum power. So, the constant-voltage system provides what might be called built-in protection against speaker

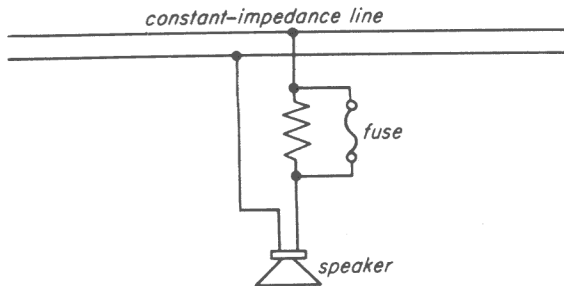


Fig. 28-23

overload. This results in a secondary advantage in the form of improved system flexibility, because it is safe to employ an amplifier capable of more power than the speaker load draws (Fig. 28-22b). If, at any time, an additional speaker has to be added to the system, this can be done without installing another amplifier. That's what we mean by flexibility. Turn back to Fig. 28-8 which is an example of a complete PA system using amplifiers rated to deliver more power than the speaker load consumes. The arrangement is safe because the constant voltage system of connecting speakers is employed.

Where the constant impedance system is employed, fuses are sometimes used to protect the speakers against overload. The fuses are the ordinary kind, but the manner of connecting them is a bit unusual (Fig. 28-23). The fuse is connected in series with the speaker just like a power fuse is connected in series with a power transformer. But, unlike the power fuse, this one has a resistor connected in parallel with it. Normally, the fuse shorts out the resistor. When an overload causes the fuse to blow, the resistor is no longer shorted out. Instead, it is effectively in series with the speaker load. Some of the power delivered by the amplifier is consumed in the resistor. The speaker operates at reduced power but is not cut out completely when the fuse blows.

28-3. ACOUSTICS

Acoustics is the study of the behavior of sound. There are several behavior characteristics of sound that are important in PA work.

These are as follows:

1. Attenuation
2. Absorption
3. Cancellation
4. Reverberation
5. Acoustic feedback

By *attenuation* we mean that sound decreases in intensity the farther you get from the source. *Absorption* refers to the fact that the presence of soft materials in a room decreases the intensity of sound in addition to the normal attenuation and makes the room sound *dead*. *Cancellation* is a decrease of sound intensity that occurs when two out-of-phase sound waves in air exist together in the exact same place. The intensities of the two sound waves cancel each other and what you hear is what remains when the weaker sound is subtracted from the stronger. *Reverberation* is the reflection of sound waves from hard surfaces. The opposite of absorption, it increases sound intensity in a room above what the level of intensity would be, if there were no reverberation, and makes the room sound *live*. *Acoustic feedback* is the regenerative feeding back of the output signal of an amplifier, back to the amplifier input. The feedback takes place not electrically but through the air from the loudspeaker back to the microphone. Although not electrical, acoustic feedback has many of the characteristics of electrical regenerative feedback. Like electrical feedback, acoustic feedback can cause oscillation and the generation of a kind of self-sustained tone usually described as a *howl*.

In general, the five acoustic qualities of sound that we have listed are important in PA work because they bear upon the placement of loudspeakers and microphones, choice of the kind and make of speaker to be used in a particular installation, and the amount of sound power required for a particular installation.

Attenuation bears upon the placement of loudspeakers because as a sound wave

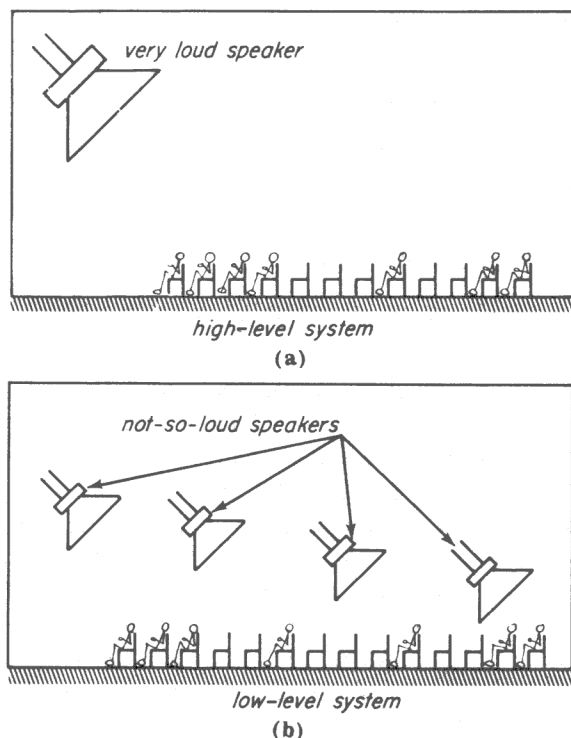


Fig. 28-24

travels away from the speaker that produced it, it gets weaker and weaker. If the sound is to be heard a long way off, it must be very loud to begin with, and this is unpleasant for people situated near the speaker (Fig. 28-24a). A way to overcome this difficulty is to use several not-so-loud speakers spaced at 30 to 40 foot intervals along the distance to be covered by the system (Fig. 28-24b). Everybody gets subjected to about the same loudness of sound.

This arrangement is called a *low-level* speaker system, since the loudness of the individual speakers is less than it would be if one very loud speaker were used. A system employing one or more very loud speakers instead of many not-so-loud speakers, is called a *high-level* system. Although a person situated very near the very loud speakers would probably find them unpleasant, the system is quite practical; the very loud speakers can often be located so that everybody being addressed is sufficiently far away not to be subjected to unpleasantly loud sound. An example is the cluster of very loud horns seen suspended on a platform over a sports arena. The choice — whether to use a high-level or a low-level speaker

system — is affected by the other acoustical conditions of an installation as well as attenuation. Where it is feasible, the high-level system is usually cheaper than a low-level system installed under the same conditions.

The greatest total volume of sound is obtained when the electrical connections to speakers is made in a way to cause the diaphragms or cones to move in and out together. This is called phasing the speakers. If all the speakers are manufactured so that they can be wired up in the same way, phasing can be accomplished by connecting the same lead from the speaker line to the same speaker terminal as shown in Fig. 28-25a, and not as shown in Fig. 28-25b. To find out if two dissimilar speakers are internally wired the same, you can momentarily touch the terminals of a 1.5-volt flashlight battery to the terminals of the speakers. Watch to see which way the cone moves. If the positive terminal of the battery is touched to the left hand terminal of both speakers but the cones move in the opposite directions, the internal wiring of the two speakers is opposite.

Phasing does not greatly affect the total output of speakers that are separated as in a low-level system. It is most important when the speakers are in a cluster. Even here, a speaker is occasionally connected out of

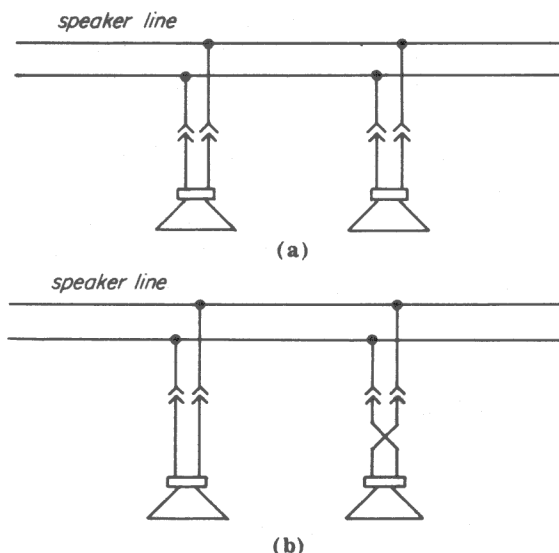


Fig. 28-25

phase to alter the directional disbursement of sound power from the cluster when some acoustic problem must be overcome. The total sound power is reduced when this is done.

As a rule of thumb, you can assume that sound gets attenuated by 75 percent to 25 percent of what it was (a 6-db decrease in sound power) every time you double the distance from a loudspeaker. If you have 10 watts of sound power 10 feet from a speaker, you can expect to have 2.5 watts 20 feet from the speaker. By watts of sound power, we mean watts of mechanical power of the sound coming out of the speaker and not watts of electrical audio power going into the speaker. Ten watts of sound power is very loud. Cone speakers can be about 2 percent efficient; it would take about 500 watts of electrical audio power to produce 10 watts of sound power with cone speakers. Horns can be 25 percent efficient; it would take 40 watts of electrical audio power to produce 10 watts of sound power with 25 percent efficient horns.

If the room in which a PA system is installed is very dead, the sound intensity will decrease at a faster rate than the 6-db rate due to normal attenuation. The additional decrease is due to absorption of the sound by soft material like drapery and upholstery in the dead room. More sound power is needed in a dead room. On the other hand, if the room is live, the sound will not decrease as much as the 6-db normal attenuation rate. This is because sound gets reinforced by returning sound energy that is reflected from hard surfaces like bare plaster walls and hardwood floors.

If you have ever inspected a newly built unfurnished house and then inspected the same house with furniture, rugs, and drapery installed, you know the difference between a live and a dead room and you have some idea of how the materials present in a room cause it to sound live or dead. Talk in an unfurnished room has a live, vibrant quality; less effort is required to talk loud. The presence of some upholstered furniture, drapes, and rugs brings about a remarkable change; the

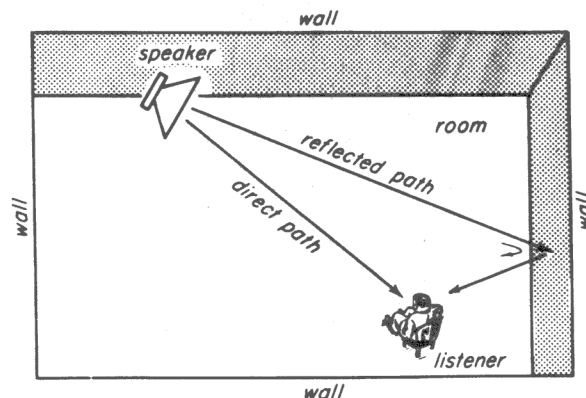


Fig. 28-26

acoustics of the room are dead. Engineers have constructed some very dead rooms for the testing of loudspeakers and microphones by heavily padding floors, walls, and ceiling. If you try to shout in such a room, you can feel the strain upon your throat; but the sound that comes out is not loud.

Normally, a completely dead room is not desirable for the reproduction of sound. Some reverberation is desirable because it gives the sound a vibrant quality. On the other hand, too much reverberation creates problems. As you know, reverberation is due to the reflection of sound from hard surfaces. Figure 28-26 shows that a longer time is required for reflected sound to reach a listener than for sound traveling directly. If the time delay is long enough, an echo is produced because the listener hears the same sound once from the direct path and a second time from the longer reflected path. The hearing of the same sound at two different times is responsible for echo and also all the other sound effects that come under the heading of reverberation. These effects are as follows:

1. When reverberation is slight, the sound takes on a pleasant vibrant quality.
2. Somewhat more reverberation causes speech to become garbled and hard to understand.
3. Still more reverberation causes both echoes and cancellation effects.

For the second and third effects of reverberation to take place, there must be a

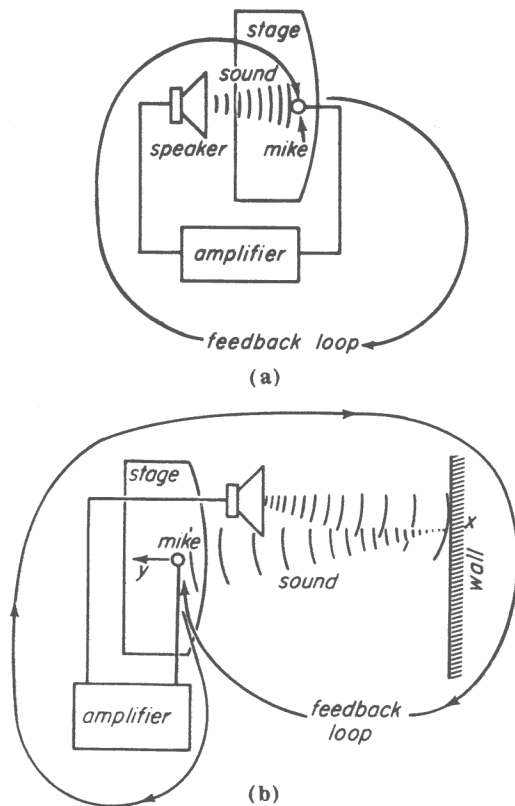


Fig. 28-27

long time lag between hearing the direct and the reflected sound. For this, the length of the reflected path must be long. Consequently, the room must be large. Reverberation does not introduce problems in PA systems installed in small rooms. It is a problem in large auditoriums, especially in large churches. Out-of-doors, about the only reverberation problem that can arise is an echo.

Cancellation takes place in large irregularly shaped auditoriums. You are likely to find a nook in such a place where the sound cannot be heard although it is loud enough in the immediately surrounding area.

Sometimes, as in a radio studio or a theater, the liveness of a room is carefully controlled during its construction by acoustic design. Normally, however, PA systems are installed in places where no attempt has been made to tailor the room acoustics to the reproduction of sound. When a reverberation problem arises, it can be solved by the judicious hanging of a drapery (usually on a wall towards which the speakers are aimed).

But more often, the problem is solved by the use of a low-level speaker system with many not-so-loud speakers spaced at intervals instead of a few very loud speakers located centrally. This works because the sound produced by each speaker is weak to begin with; after it travels the long reflected path, it is further weakened by attenuation to a degree that it is not troublesome. Speakers in a low-level system should not be spaced more than 40 feet apart or the sound reaching a listener from two different speakers will be delayed sufficiently to cause bad effects of the same kind that are caused by reverberations. It is the inappropriate placement of speakers that causes most echoes encountered in outdoor PA installations.

Another sound effect that has to do with reverberation is acoustic feedback. Acoustic feedback has to do with reverberation in that sound may be fed from speaker back to microphone via a reflected path and so be caused by reverberation. However, acoustic feedback may also take place via a direct path and so it is independent of reverberation. Figure 28-27a shows an acoustic-feedback loop taking a direct path. You see, it is not a good idea to mount speakers behind a microphone. Doing so allows acoustic feedback to take place via a direct path. When acoustic feedback is via a direct path, the strength of signal fed back is great, and the system howls at a low setting of the volume control. The strength of signal feedback via a reflected path (shown in Fig. 28-27b, where the speaker is in front of the mike), is not as great as via a direct path; the howl does not start until the volume is advanced to a high setting. You can understand that a drapery hung on the wall at point X will reduce the strength of signal fed back and thus alleviate the howling. Also, use of a directional microphone that responds best in the direction of the arrow at Y will reduce feedback as will rotation of the loudspeaker so that it points away from the wall.

Acoustic feedback limits the maximum power at which a PA system can operate to a value below the rated maximum power capability of the system. Sometimes this is

an acceptable condition, sometimes not. For example, a system might howl when turned up to overcome the noise made by conversation in an audience, but, if the people stop talking when they hear the howl and give their attention to the sound from the speaker, less power will be required to address them. The system can then be turned down and the howl stopped. This is an example of an acceptable condition.

Acoustic feedback is greater at low audio frequencies than at high audio frequencies. Sometimes, the howl can be stopped by reducing the low frequency response of the amplifier by means of the tone control.

28-4. SETTING UP A PA SYSTEM - AMBIENT NOISE

Ambient means surrounding. Ambient noise is noise outside or surrounding an electrical system like a PA system, as distinguished from the noise that is produced by the system itself. Ambient noise is the noise that is naturally present in the place where a PA system is to be set up. The need of overcoming ambient noise is one of the principle reasons for using a PA system. In other words, one reason why PA systems are needed is to increase the the natural loudness of a speaking voice or other sound to a point where it is louder than surrounding noise. When this is done, the speaking voice or sound can be heard by people who were not able to hear it without amplification because the ambient noise was loud compared to the natural loudness of the speaking voice or other sound. One thing that must be considered when setting up a PA system is the level of ambient noise to be overcome.

Another reason for the use of PA systems is to increase the natural loudness of a sound to the degree that it can be heard by people too far away to hear the sound at its natural loudness level. To determine how much sound intensity is needed in order for it to be heard by all people within a given

area, both ambient noise and the size of the area have to be taken into consideration.

When an enclosed area, such as a room or auditorium is to be covered, the natural attenuation rate of sound intensity that takes place as the sound waves travel over the area is either lessened by reverberation from the sides of the enclosure or increased by absorption. These factors have to be considered when planning an indoor PA system.

Ambient noise is not equally intense at all audio frequencies. It is most intense at low audio frequencies. So the intensity of high-pitched sound that is required to overcome ambient noise is less than the intensity of low-pitched sound required to overcome the same amount of noise. Fortunately, the high-pitched sounds are the ones that cause the words to be easily recognized. This is fortunate because where only speech is amplified in a PA system, less amplification is required to make the speech recognizable in the presence of the ambient noise. This is where horns have an advantage over cone-type speakers. Horns which do not reproduce low frequencies at all can be used to reproduce speech. Because this type of horn can be small, it is not expensive. Because horns are efficient, little electrical audio power is required to produce the sound power needed to overcome the ambient noise. An example of a PA system intended to amplify only speech sounds is a paging system used in an airlines terminal. The speakers shown in Fig. 28-16 are designed for paging applications.

For a faithful reproduction of music, the low audio frequencies must be reproduced. So when a PA system is intended to reproduce music, more sound intensity must be built up to overcome the ambient noise. However, as a practical matter, the amount of low-frequency sound power required to overcome the ambient noise is sometimes so large that the cost of equipment to produce it is not warranted. And unless enough low-frequency sound power is produced to overcome the ambient noise, it is useless to produce any low-frequency sound power at

all. What you hear will be the same whether or not you produce low-frequency power as long as you do not produce enough to overcome the noise. A properly set up PA system in a noisy factory intended to serve mostly for paging purposes but also to occasionally reproduce recorded music may not be equipped to produce low-frequency sound power.

Now we are in a position to outline some of the steps that must be taken toward setting up an indoor PA system. These are as follows:

1. Determine the ambient noise intensity.
2. Measure the volume of the room to be covered.
3. Determine how much sound intensity you will need to overcome the ambient noise.
4. Correct this value of sound intensity to a lesser value if the room is very live or to a larger value if the room is very dead.
5. Again correct the value of needed sound intensity. Increase the value if you want to reproduce music and decrease the intensity value if you want to reproduce only speech.
6. Now that you know how much sound intensity you need, choose a speaker that can develop an acoustic power great enough to produce the required sound intensity in a space equal to the volume of the room (measured in Step 2). On the basis of the speaker manufacturer's specifications, decide how much electrical audio power must be supplied to the speakers in order for them to produce the desired sound power output.
7. Choose an amplifier capable of delivering the needed electrical audio power.
8. Arrange the equipment for the greatest convenience; i.e., to accommodate a convenient number of mikes and phonographs conveniently located, etc.

Obviously, to fulfill the requirement in Step 1, there must be a method of measuring sound intensity — in this case, noise in-

tensity. There are instruments for measuring the intensity of sound. One is called *The American Standard Sound Level Meter*. However, for the purpose of setting up a PA system, it is not necessary to be very accurate in determining the intensity levels of noise and wanted sound. It is enough to estimate the intensity, allowing a safety factor to cover errors of approximation. When a run-of-the-mill PA system is being planned the practice is to determine noise level (Step 1) by consulting published charts that show the noise levels usually present in certain places under certain conditions. For example, the chart might show the level of noise intensity for a noisy factory. Although intensity levels are sometimes measured, it is common practice to rely upon charts. As a matter of fact, accurate measurement is difficult.

Whether charts or actual measurements are used, there must be a unit of measure for gauging sound intensity. Sound may be measured in units of intensity and units of pressure. The *intensity level* of a sound is given in db. Naturally, when we use db, there has to be a reference. The reference is called the *reference intensity*. Unless there is a specification that tells you that the reference intensity is something else, you can assume that the reference intensity is 10^{-16} watt per square centimeter. The formula for intensity level is:

$$\text{db} = 10 \log \frac{I}{I_0}$$

where:

I = intensity of sound in watts per square centimeter

I_0 = 10^{-16} watt per square centimeter

The *pressure level* of sound is also given in db. The reference for pressure level is called the *reference pressure*. Unless you have reason to believe otherwise, you can assume that the reference pressure is 0.0002 dyne per square centimeter. The formula for

pressure level is:

$$\text{db} = 20 \log \frac{P}{P_0}$$

where:

P = pressure of sound in dynes per square centimeter

P_0 = 0.0002 dyne per square centimeter

Since it is possible to express intensity level and pressure level in decibels, it is possible to use simple arithmetic for calculating relationships in which these terms are important. When either intensity level or the pressure level that produces it is expressed in db, the number of db is the same; for example, 10 db of pressure produces 10 db of intensity.

For instance, Step 3 of the procedure we have just described requires that you determine the sound intensity needed to overcome the ambient noise. The intensity required depends upon how intense the ambient unwanted noise is. Under normal conditions, ambient noise must be exceeded by 5 db. Thus, the way to compute the intensity needed to overcome noise is to add 5 db to the number of db of unwanted noise. This is a very simple system, and you won't go wrong so long as you do all your calculations in db units.

We have said that you must add 5 db of intensity to the noise intensity level in order to overcome normal noise. However, if the noise is abnormally high, 5 db is too small a proportion of the total noise in db to have any effect. It is necessary to add 10 db of intensity level to overcome high noise.

Step 6 requires that you find out the acoustical power that will produce the desired sound intensity in the volume of space to be covered. This information is available in published charts. Acoustic power (not

intensity) is measured in watts or dbm (db to the reference 10^{-3} watts). After this, you must find a speaker or speakers that can produce this power. Then you must find out how much electrical power has to be furnished to each speaker to produce the desired sound power. Doing this is complicated by the fact that various speaker manufacturers use various systems of rating their speakers.

University Loudspeakers, Inc. manufactures speakers for use in PA systems and publishes a chart that combines much valuable information. The chart is reproduced in Fig. 28-28.

Steps 1 through 7 can be carried out with this chart to find a suitable University Speaker for a particular PA installation and the amplifier power output that will drive it.

Let us use this chart to go through the steps of setting up a PA system. Also, let us examine the chart. Near the right-hand side, there is a column labeled *Background Noise Level in db*. This column gives the ambient noise level in db (to reference 10^{-16} watt/cm²) found in various places (listed in the column labeled *Average Specific Application*) such as a church (55 db) and a very noisy factory (90 db). At the far left-hand side (on the Y axis of the graph) is a corresponding scale of db to the reference 10^{-16} watt/cm². This scale (labeled *AVERAGE SOUND LEVEL IN DB DELIVERED BY SOUND SYSTEM*) shows the level of sound intensity that will overcome the noise intensities on the right-hand scale. Notice that a noise level of 55 db shown on the right-hand scale can be overcome by a 60 db intensity of wanted sound. Remember, we mentioned that normal noise could be overcome by a 5 db advantage of wanted sound. A 90 db noise level (abnormally high) requires 100 db of wanted sound intensity, (a 10 db advantage).

The horizontal scale of decibels located between the large upper graph and the small lower graph shows electric input power in dbm. The slant lines on the large graph each represent a certain room volume. To find the amount of electric power that a speaker

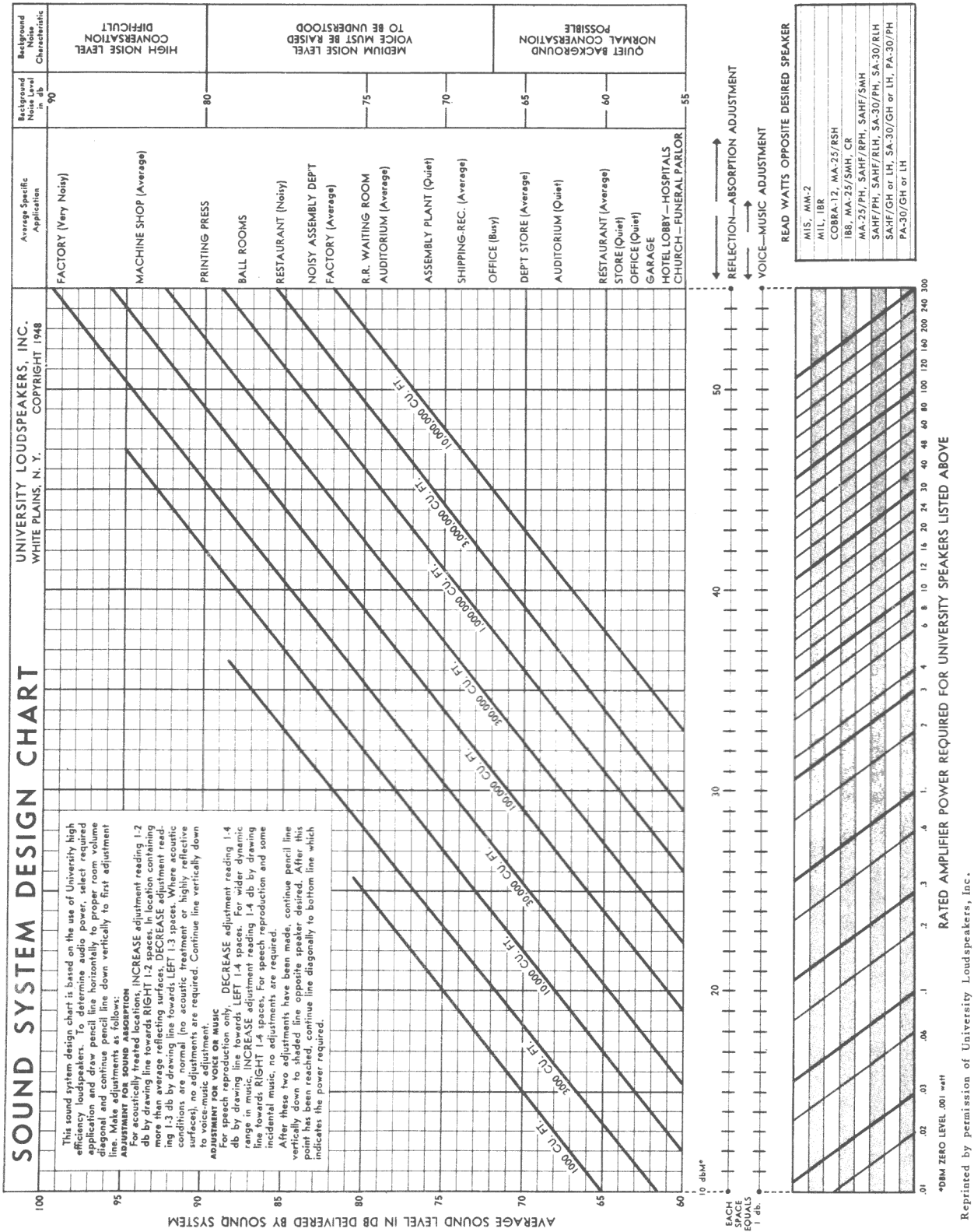


Fig. 28-28

must receive to produce a wanted sound intensity that can overcome the ambient noise in a room of a certain size, you can proceed as follows. Suppose you want to install a PA system in a 1,000,000 cubic-foot railroad waiting room. Looking up the railroad waiting room in the right-hand column, you find that the noise level is 75 db. That the noise can be overcome by 80 db is shown in the left-hand column. So one way in which you can overcome the ambient noise is by furnishing a sound level of 80 db.

However, you can use the graph in a different way. You can draw a horizontal line from the 75 db railroad waiting room until it intersects the slanting line that indicates 1,000,000 cubic feet. From this intersection, you can draw a vertical line down to the *REFLECTION - ABSORPTION ADJUSTMENT* scale. To correct for the fact that the railroad waiting room is a live room, you move from 1 to 3 spaces left along this scale. Then you draw a line straight down to the *VOICE-MUSIC ADJUSTMENT* scale. Since only speech is to be reproduced, you move 1 to 4 more spaces to the left. Let us suppose that you move three spaces to the left along each scale — a total of six spaces.

At this point, if you extend the line vertically back up to the *REFLECTION - ABSORPTION ADJUSTMENT* scale, and down to the very bottom scale, you can find the number of dbm of audio power input required and the actual number of watts that corresponds to this number of dbm. For example, a line drawn down from 40 dbm on the *REFLECTION - ABSORPTION ADJUSTMENT* scale intersects the 10-watt point on the bottom scale. As you know, 10 watts is equal to 40 dbm.

Let's stop here and consider what we have found out. We have learned that for a particular level of ambient noise in a room of given area, a particular sound level is required. For instance, in the example we chose, we found that 80 db was required to overcome 75 db of ambient noise.

However, we can use this chart to select a University Loudspeaker without considering the sound level required to overcome ambient noise. Note that this chart cannot be used for the speakers of other manufacturers; it has been designed to be used only for University speakers.

We were able to find out the electric power in dbm, and, consequently, the number of watts of input required to overcome a particular ambient noise. In our example, in which we are considering how to overcome an ambient noise level of 75 db in a railroad station space of 1,000,000 cubic feet, we find that an input power of 40 dbm, or 10 watts must be supplied to the most efficient of the University Loudspeaker, Inc. speakers in order to overcome the ambient noise under the conditions we described.

Earlier in this lesson, we covered some of the requirements that should guide you in your choice of a speaker.

If you are using the chart in Fig. 28-28 to choose a speaker made by the manufacturer who supplied the chart, you would choose from among the speakers listed in the box in the lower right-hand corner of the chart. Each of these speakers has particular characteristics, which you could find by looking up the speakers in the manufacturer's catalog. After you had found a speaker that you considered suitable for your needs, you could determine the relative efficiency of that speaker with respect to the other speakers listed on the chart by referring to the chart.

Suppose that you decided that the PA-30/GH speaker was best suited to your requirements, and that you were going to use that speaker in the situation we have been discussing — that is, you were going to use the speaker to overcome 75 db of ambient noise in a volume of 1,000,000 cubic feet.

If you draw a line connecting the 40 db mark with the 10-watt mark, this line goes through the shaded bar opposite the PA-30/GH loudspeaker.

According to the directions, when you reach the shaded (or unshaded, however the case may be) bar opposite the speaker that you have chosen, you are to continue the line diagonally.

If you happen to hit one of the diagonal lines already drawn just at the point where it enters the bar opposite the speaker that you are interested in, you follow this diagonal line to its end. In our case, the line we draw leads us to a point on the bottom of the shaded bar, just where the slanted line intersects 10 watts. Therefore, we find that the input required for the speaker is 10 watts. When 10 watts is supplied to this speaker, it will overcome the ambient noise we have been discussing.

Had we chosen to use the MIS speaker, which we might very well have done because of certain factors, we would have found that an input of about 27 watts was required to overcome the same ambient noise. Therefore, the efficiency of this speaker would be relatively less than that of the PA-30/GH. But, as we have already pointed out, efficiency is only one among many factors to consider.

This system is handy for setting up indoor PA systems. The procedure is somewhat different for setting up an outdoor PA installation where there are no walls to cause reverberation and thus reinforce the sound intensity. The exact procedure depends upon what kind of system the manufacturer has used to rate his speakers. We have illustrated a system used by University Loudspeakers, Inc.

There is a speaker-rating system which gives the sound intensity that the speaker will produce at a point 30 feet directly in front of the speaker (out-of-doors) when a specified electrical power input is applied. Suppose that a speaker is rated to produce a sound intensity level of 100 db to the reference 10^{-16} watt/cm² with 25 watts input electrical power. Since sound that is unaffected by reverberation gets attenuated 6db every time the distance from the speaker

is doubled, the intensity will be 100 db minus 12 db or 88 db at 120 feet from the speaker. Following this principle, you can aim the speaker at the most distant point to be covered and calculate the intensity level that the speaker will produce at this distance. If the speaker will produce an intensity great enough to overcome ambient noise at the most distant point to be covered, it will also overcome noise at closer points. Of course, you must know the angle of dispersion of the speaker to know how wide an area you will cover. This angle is usually specified by manufacturers of horns. For cone speakers, it is 180 degrees up to 500 cps (the audio frequencies that account for most of the total power in sound). You must also be sure that the intensity level produced by the speaker is not unpleasantly loud for people nearest the speaker. A level of 120 db to the reference 10^{-16} watts/cm² is too loud for comfort. A level of 100 db to the reference 10^{-16} watts/cm² can overcome about the loudest noise that it is practical to try to overcome. Just as the sound gets attenuated 6 db every time you double the distance from the speaker, it becomes 6 db more intense every time you halve the distance. In our example, intensity is given as 100 db at 30 feet in front of the speaker. The intensity is 106 db at 15 feet in front of the speaker.

Another useful rule-of-thumb applicable out-of-doors is that each time you double or halve the electrical power input to a speaker, you also double or halve the sound intensity that the speaker produces. Naturally, the rule does not hold above the maximum input-power handling capability of the speaker. Doubling or halving an intensity level given in decibels to the 10^{-16} watt/cm² reference changes the numerical value of the intensity level by ± 3 db. In our example, reducing the input power from 25 to 12.5 watts reduces the intensity from 100 to 97 db.

Needless to say, experienced PA men work out their own shortcuts for simplifying these procedures, such as learning from experience how much noise a horn of a certain size and shape will overcome at a

certain distance or in a certain size room and with a certain electrical power input to the horn.

28-5. USING A TAPE RECORDER WITH A PA SYSTEM

Tape recorders have been used in conjunction with PA systems since these recorders became popular at the close of World War II. Both professional and home-type tape recorders are used with all kinds of PA systems. Recorders are used both to record from a PA system and to play back into a PA system. We will limit our brief discussion to home-type machines and to the recording function, since play-back operation simply involves plugging the machine into the phono input of a PA system.

Suppose a PA system is in use in an auditorium for the purpose of addressing a meeting of a trade association. You want to record the proceedings with a tape recorder. The simplest approach is to set up the machine independent of the PA system — just connect a mike to the recorder and record. This approach is simple, but it is not satisfactory. It is not possible with a single uncontrolled mike to make a recording of the sounds made by even a few people who are separated from each other and who talk and make sounds at random and sometimes simultaneously. To make such a recording properly requires the use of several mikes carefully located. The output of each mike must be monitored and controlled. A preamplifier-mixer circuit is used for this purpose. This circuit can be controlled either manually or automatically. But either method involves the use of professional recording equipment. When the recording is made by simply plugging a mike into a tape recorder and recording, it sounds, on playback, like a meaningless noise. Individual sounds cannot be distinguished. The words spoken by individuals cannot be understood.

However, if instead of setting up the tape recorder independently, you connect it to the PA system, a clear recording will be obtained. The reason for this is that the

sound signal present in the PA system is, by virtue of the nature of PA systems, always at a level above that of the ambient noise. In our example — the trade-association meeting — the sound signal present in the PA system represents speech put directly into the microphones associated with the system. It does not include side comments made by members. These sounds are part of the ambient noise, which gets drowned out. If part of the proceedings at the meeting involves discussion on the part of the association members who are in the audience, it will be necessary for each participant in the discussion to talk into a microphone in order to record his words.

There are a few rudimentary ideas that should be understood concerning the connection of tape recorders to a PA system. Two recorders are usually connected simultaneously. The tape is set in motion on one machine only and recording on this machine only continues until the reel of tape is almost exhausted. Then as the reel on the first machine plays out, the second machine is set in motion so that the start of the second reel and end of the first reel get recorded simultaneously. This method avoids missing any of the material to be recorded. Some material would be missed during the time required to change reels if only one machine were used. As the second machine records, the reel is changed on the first machine preparatory to repeating the changeover process when the reel on the second machine is used up.

In order to electrically connect the two machines simultaneously to the PA circuit, isolating resistors must be used. The resistors can be mounted in a junction box as shown in Fig. 28-29. This arrangement is suitable for connecting the recorders to a single ended speaker line of a PA system (Fig. 28-30). The phono input terminals of the tape recorders should be used because the signal voltage level on PA speaker lines is more than high enough to drive a tape recorder from the phono input. Of course, the volume control on the tape recorder can be adjusted to set the signal

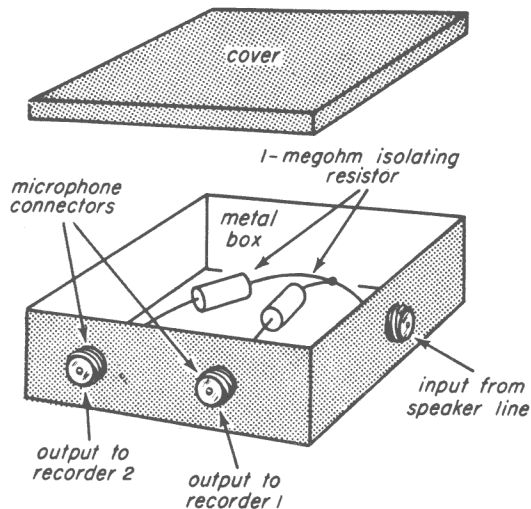


Fig. 28-29

voltage at the proper level for recording. Connecting the phono-input terminal of a home-type recorder to a speaker line grounds one wire of the line. If the line is balanced, this upsets the balance, halves the line impedance, and mismatches the amplifier.

As one machine records, the other is standing by, electrically connected to the PA system. If you preset the volume control on the stand by machine, you may encounter acoustic feedback in the PA system. Most home-type tape recorders are arranged so that when they are tuned on but not recording, any signal applied to the input of the machine is amplified and reproduced in the recorder's speaker. (The recorder can be used as a small PA system in this mode of operation.)

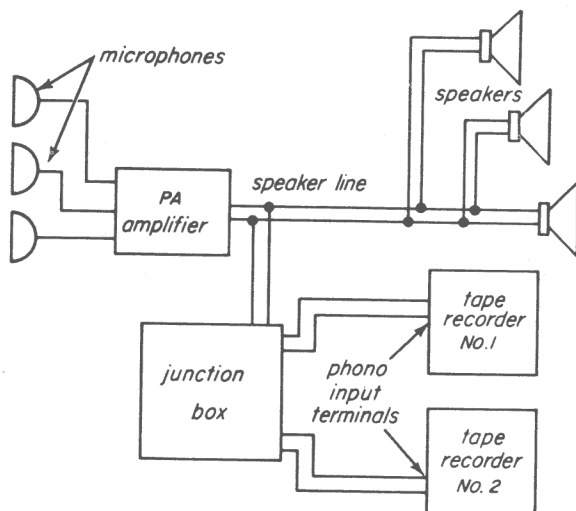


Fig. 28-30

Because of this feature, acoustic feedback can take place via a path from the recorder speaker to the PA microphone. Feedback can be prevented by setting the recorder volume control at minimum, but it is convenient to have the control preset at the proper recording level. The problem is how to shut off the recorder speaker without turning down the volume. Often this can be accomplished by inserting a blank (unwired) plug in the external speaker jack on the recorder. Usually, the recorders are wired so that when a plug is in place in the external speaker jack, the internal speaker is inoperative.

28-6. MAINTENANCE

The procedure for troubleshooting PA amplifiers is the same as that followed in troubleshooting radio audio circuits. The inter-connecting cables used in a PA installation can be tested for shorts and opens with an ohmmeter. These continuity tests are the same as described in previous booklets. While the troubleshooting methods used to service PA systems are like the methods used with radio receiver circuits, there are two ideas involved in PA work that do not come up in connection with radio service work. These are the emergency repair and a system of preventive maintenance.

PA systems are used in situations where a breakdown can have serious consequences, such as a paging-announcing system in an airline terminal or a PA installation in a theater. *Special precautions must be taken to minimize the time during which the PA system is inoperative during maintenance.* For this reason, it is frequently necessary to make a temporary repair, often called an emergency repair, that will serve to keep a PA system operating during the time required to permanently correct the defect. The temporary measure taken may be as simple as short circuiting an on-off switch so that the system will operate while the new on-off switch is on order. Or it may be necessary to temporarily install a spare amplifier (perhaps a larger one than the permanent unit) while the permanent amplifier is in

the shop for repair. Again, it may be necessary to install an exposed cable temporarily while the permanent cable, which might be concealed in a wall or buried underground, is replaced.

Preventive maintenance is a system sometimes used by PA service organizations to forestall breakdowns of the PA systems. It provides for the periodic inspection and repair of PA installations before they break down. Such methods are widely used in the maintenance of electronic equipment that must not fail, such as broadcast radio transmitters and radio equipment carried in aircraft. We will outline a PA inspection to show what the serviceman looks for and does.

When he makes his inspection, the serviceman asks the people who use the PA system if they have noticed any faulty operation. Bearing their comments in mind, he then operates the equipment himself. He looks out for the following kind of minor defect which may indicate more serious trouble to come.

1. Excessive hum, which may precede the complete failure of a power-supply filter capacitor. Excessive hum may be due to an open shield, which is an indication of a cable that is so worn that it will break apart soon.

2. Excessive distortion, which may indicate the presence of a leaky (partially shorted) capacitor in an amplifier. In time,

the capacitor may become completely shorted and make the amplifier completely inoperative.

3. Slow warmup or weak output which may indicate weak tubes.

4. Intermittently operating controls which will eventually fail completely.

The serviceman either tests all tubes with a tube tester or replaces them as a matter of course after they have been working a certain length of time. Sometimes, all voltage amplifier tubes are checked while the power amplifier tubes and the power supply rectifier tubes are periodically replaced.

Tubes have a long shelf life. That is, they can be stored a long time and will remain in good condition. But they gradually get used up in operation. A curious fact, worth noting is that many other electronic component parts can become defective in storage as easily as in operation. For example, the wire in a coil can corrode and break while the coil is in storage as well as while the coil is in use. Electrolytic capacitors can dry up on the shelf.

Another thing the serviceman does during routine inspection is visually inspect all cables and connectors for signs of wear which indicates that a break will occur eventually.
